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TRANSMITTAL OF APPEAL BRIEF (Large Entity)	Docket No. 02100033AA

In Re Application Of: K. Ozawa

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Application No.	Filing Date	Examiner	Customer No.	Group Art Unit	Confirmation No.
09/302,397	April 30, 1999	A. A. Armstrong	30743	2654	6830

Invention: SPEECH CODING APPARATUS AND SPEECH DECODING APPARATUS

COMMISSIONER FOR PATENTS:

Transmitted herewith in triplicate is the Appeal Brief in this application, with respect to the Notice of Appeal filed on

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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

BEFORE THE BOARD OF PATENT APPEALS AND INTERFERENCES

In re patent application of

Kazunori Ozawa

Serial No. 09/302,397 — Group Art Unit 2654

Filed April 30, 1999 — Examiner Angela A. Armstrong

For SPEECH CODING APPARATUS AND SPEECH DECODING APPARATUS — Confirmation No. 6830

Commissioner for Patents
PO Box 1450
Alexandria, Virginia 22313-1450

APPELLANT'S BRIEF UNDER 37 C.F.R. §1.192

This brief, which is filed herewith in triplicate, is in furtherance of the Notice of Appeal, filed in this case on July 23, 2004.

This brief contains these items under the following headings, and in the order set forth below (37 C.F.R. §1.192(c)):

I. REAL PARTY IN INTEREST

The real party in interest in the appeal is:

☐ the party named in the caption of this brief.

☑ the following party: NEC Corporation of Tokyo, Japan

II. Related Appeals and Interferences

With respect to other appeals or interferences that will directly affect, or be directly affected by, or have a bearing on the Board's decision in this appeal:

☑ there are no such appeals or interferences.

☐ these are as follows:

III. STATUS OF CLAIMS

The status of the claims in this application are:

A. Total number of claims in Application

Claims in the application are: claims 1 to 11

B. Status of all the claims:

1. Claims cancelled: claim 5
2. Claims withdrawn from consideration but not cancelled: none
3. Claims pending: claims 1 to 4 and 6 to 11
4. Claims allowed: none
5. Claims rejected: claims 1 to 4 and 6 to 11

C. Claims on Appeal.

The claims on appeal are: claims 1 to 4 and 6 to 11

IV. Status of Amendments

The status of amendments filed subsequent to the final rejection are as follows: A Request for Reconsideration was filed on June 28, 2004, together with a Petition for extension of time and fee in response to the Final rejection mailed on January 28, 2004. An Advisory Action was mailed on August 9, 2004, indicating that the Request for Reconsideration had been considered but did not place the application in condition for allowance.

V. SUMMARY OF INVENTION

The invention as defined in the claims on appeal is directed to a speech coding apparatus for coding a speech signal at a low bit rate with high quality. The invention effectively suppresses deterioration in sound quality in terms of background noise while minimizing the calculations required.

The claimed invention provides a speech coding system which succeeds in reducing the required calculations while at the same time maintaining good sound quality in terms of background noise for low bit rates. Four embodiments of the speech coding system are shown in Figures 1 to 4, respectively, and a speech decoding system is shown in Figure 5.

With reference to Figure 1 of the drawings, the invention employs a spectrum parameter calculation section which includes a spectrum parameter calculation circuit 200 for extracting spectrum parameter from a speech signal and a spectrum quantizing circuit 210 for quantizing the spectrum parameter. An adaptive codebook section includes adaptive codebook 500 and sound source quantization circuit 350. A mode discrimination circuit 370 discriminates the mode on the basis of the past quantized gain from gain quantization circuit 366. The mode discrimination circuit 370 receives the adaptive codebook gain quantized by the gain quantization circuit 366 one subframe ahead of the current subframe and compares it to a predetermined threshold to perform voiced/unvoiced determination. When a predetermined mode is discriminated, a sound source quantization circuit 350 searches combinations of code vectors stored in a sound source code books 351 or 352, which are used to collectively quantize the amplitudes or polarities of a plurality of pulses, and a plurality of shift amounts used to temporally shift predetermined pulse positions, to select a combination of a code vector and shift amount which minimizes distortion relative to input speech. Codebook 351 is used for voiced sound, and codebook 352 is used for unvoiced sound. Appellant uses separate codebooks for voiced and unvoiced

speech signals so as to minimize calculations without adversely affecting sound quality in terms of background noise. A multiplexer section 400 outputs a combination of an output from the spectrum parameter calculation section 210, an output from the adaptive codebook section 500, and an output from the sound source quantization section 350.

The speech decoding apparatus of the invention is shown in Figure 5 and includes a demultiplexer section 510 for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information. A mode discrimination section 530 discriminates the mode on the basis of the past quantized gain of the adaptive codebook. A sound decoding section 540 reconstructs a sound source signal by generating non-zero pulses from the quantized sound source information. A speech signal is reproduced or resynthesized by passing the sound source signal through a synthesis filter 560 defined by spectrum parameters.

The four embodiments of the invention represented by Figures 1 to 4 differ in the operation of the sound source quantization section. In the first embodiment shown in Figure 1, the sound source quantization circuit 350 reads out polarity code vectors from the codebook 351, assigns positions to the respective code vectors, and selects a combination of code vector and a position which minimizes distortion given by equation (11) on page 24 of the specification. For unvoiced sounds, pulse positions are set at predetermined intervals, and shift amounts for shifting the positions of all pulses are determined in advance. In this case, the sound source quantization circuit 350 receives polarity code vectors from the polarity codebook 352, and searches combinations of all shift amounts and all code vectors to select a combination of a shift amount and a code vector which minimizes distortion given by equation (15) on page 26 of the specification. The gain quantization circuit 366 receives the amplitude and position information from the sound source quantization circuit 350, receives the

voiced/unvoiced discrimination information from the mode discrimination circuit 370, reads out gain code vectors from the gain codebook 380, and selects one gain code vector that minimizes equation (16) on page 28 of the specification for the selected amplitude code vector or polarity code vector and the position.

The second embodiment of the coding system according to the invention shown in Figure 2 differs from the first embodiment shown in Figure 1 in the operation of a sound source quantization circuit 355. When voiced/unvoiced discrimination indicates an unvoiced sound, the positions are generated in advance in accordance with a predetermined rule. A random number generating circuit 600 is used to generate a predetermined number of pulse positions which are output to the sound source quantization circuit 355. If the discrimination information from mode discrimination circuit 370 indicates a voiced sound, the sound source quantization circuit 355 operates in the same manner as the sound source quantization circuit 350 in Figure 1. If, on the other hand, the information indicates an unvoiced sound, the amplitudes or polarities of pulses are collectively quantized by using a sound source codebook 352 in correspondence with the positions output from the random number generating circuit 600.

In the third embodiment of the coding system according to the invention shown in Figure 3, when voiced/unvoiced discrimination information from the mode discrimination circuit 370 indicates an unvoiced sound, a sound source quantization circuit 356 calculates the distortions given by equations (21) on page 31 of the specification in correspondence with all the combinations of all the code vectors in a sound source codebook 352 and the shift amounts of pulse positions, selects a plurality of combinations in the order which minimizes distortions, and outputs them to the gain quantization circuit 366. The gain quantization circuit 366 quantizes gains for a plurality of sets of outputs from the sound source quantization circuit 356 by using a codebook 380 and selects a combination of a shift amount, sound source code

vector, and gain code vector which minimizes distortions.

In the fourth embodiment of the coding system according to the invention shown in Figure 4, when voiced/unvoiced discrimination information from the mode discrimination circuit 370 indicates an unvoiced sound, a sound source quantization circuit 357 collectively quantizes the amplitudes or polarities of pulses for the pulse positions generated by a random number generating circuit 600 by using a sound source codebook 352 and outputs all the code vectors or a plurality of code vector candidates to a gain quantization circuit 367. The gain quantization circuit 367 quantizes gains for the respective candidates output from the sound source quantization circuit 357 by using a gain codebook 380 and outputs a combination of a code vector and gain code vector which minimizes distortion.

A main feature of the present invention as recited in the claims is that a speech coding apparatus comprises a sound source quantization section which has a codebook 351, 352 for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from the discrimination section indicates a predetermined mode, and searches combinations of code vectors stored in the codebook 351, 352 and a plurality of shift amounts used to shift positions of the pulse so as to output a combination of a code vector and shift amount which minimizes distortion.

VI. Issues

The sole issue on appeal is whether claims 1 to 4 and 6 to 11 are unpatentable over U.S. Patent No. 5,704,003 to Kleijn et al. in view of Ozawa et al., "M-LCELP Speech Coding at 4KBPS", *1994 IEEE International Conference on Acoustics, Speech, and Signal Processing*, 1994, ICASSP-94, Vol. 1, 19–22 April 1994, pp. I/269–I/272 as obvious under 35 U.S.C. §103(a).

VII. GROUPING OF CLAIMS

Group 1 includes claims 1 and 6. Claims 1 and 6 are respectively directed to the first embodiment of the coding apparatus shown in Figure 1 and the combination of the first embodiment of the coding apparatus of Figure 1 and the decoding apparatus shown in Figure 5.

Group 2 includes claims 2, 7, 9 and 10. Claims 2 and 7 are respectively directed to the second embodiment of the coding apparatus shown in Figure 2 and the combination of the second embodiment of the coding apparatus of Figure 2 and the decoding apparatus shown in Figure 5. Claims 9 and 10, dependent on claim 8, are generic to the second and fourth embodiments shown in Figures 2 and 4, respectively.

Group 3 includes claims 3 and 11. Claim 3 is directed to the third embodiment of the coding apparatus shown in Figure 3. Claim 11, which is dependent on claim 8, is generic to the embodiments of Figures 3 and 4.

Group 4 includes claims 4, 9, 10, and 11. Claim 4 is directed to the fourth embodiment of the coding apparatus shown in Figure 4. Claims 9 and 10, dependent on claim 8, are generic to the second and fourth embodiments shown in Figures 2 and 4, respectively. Claim 11, which is dependent on claim 8, is generic to the embodiments of Figures 3 and 4.

Group 5 includes claim 8. Claim 8 is generic to the several embodiments shown in Figures 1, 2, 3, and 4.

The claims do not stand or fall together. Reasons as to why the grouped claims are separately patentable are included in the arguments.

ARGUMENT VIIIA. REJECTIONS UNDER 35 U.S.C. §112, FIRST PARAGRAPH

There are no rejections under 35 U.S.C. §112, first paragraph.

ARGUMENT VIIIB. REJECTIONS UNDER 35 U.S.C. §112, SECOND PARAGRAPH

There are no rejections under 35 U.S.C. §112, second paragraph.

Argument VIIIC. Rejections Under 35 U.S.C. §102

There are no rejections under 35 U.S.C. §102.

ARGUMENT VIIID. REJECTIONS UNDER 35 U.S.C. §103

In the Final rejection mailed January 28, 2004, the Examiner rejected claims 1 to 4 and 6 to 11 under 35 U.S.C. §103(a) as being unpatentable over U.S. Patent No. 5,704,003 to Kleijn et al. in view of Ozawa et al., "M-LCELP Speech Coding at 4KBPS", *1994 IEEE International Conference on Acoustics, Speech, and Signal Processing*, 1994, ICASSP-94, Vol. 1, 19–22 April 1994, pp. I/269–I/272. The first named author of the Ozawa et al. publication is the named inventor in this patent application.

The primary reference to Kleijn et al. discloses a method of speech coding using Relaxation Code-Excited Linear Predictive (RCELP) techniques which provides a peak-to-average ratio criterion that determines whether or not time shifting of a speech residual signal should be applied within a certain sub-frame. The coders of Kleijn et al. have a characteristic feature of finding a residual signal r(n) from an input speech signal 101, and coding the residual signal r(n) by applying a time shift to the residual signal r(n) with a time warping device and delay line 107. More specifically, a time shift T, which can minimize a differential electric power between an electric power of a signal r(n-T) having a time shift T from the residual signal r(n) and that of a delayed residual signal r(n-D(n)) is firstly determined, and then a coding parameter required for the coding is extracted after applying a time shift T to the residual signal r(n).

The article by Ozawa et al. describes an M-LCELP (Multi-mode Learned Code Excited LPC) speech coder, which was developed for the North American half-rate digital cellular systems. The M-LCELP speech coder of Ozawa et al. provides multi-mode and multi-codebook coding, pitch lag differential coding with pitch tracking, a two-stage joint design regular-pulse codebook with common phase structure in voiced frames, an efficient vector quantization for LSP parameters, and an adaptive MA type comb filter to suppress excitation signal inter-harmonic noise. The

M-LCELP encoder and decoder structure is shown in Fig. 1 of Ozawa et al.

The Examiner first addressed claims 1 and 6, the Group 1 claims listed on page 12 of this brief. Claim 1 is directed to the embodiments of the speech encoder shown in Figures 1 to 4, while claim 6 is directed to the combination of a coding/decoding apparatus employing the coding system of one of Figure 1 and the decoding system of Figure 5.

Claim 1 recites, *inter alia*, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". This sound source quantization section comprises, "a discrimination section [**370**] for discriminating a voiced sound mode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook [**500**] ", and "a sound source quantization section [**350**] which has a codebook [**351, 352**] for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination circuit section, and searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech". Similar limitations are recited in claims 2 and 3. Claim 6 recites "a demultiplexer section [**510**] for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information", "a mode discrimination section [**530**] for discriminating a mode by using a past quantized gain in said adaptive codebook", "a sound source signal reconstructing section [**540**] for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information when an output from said discrimination indicates a predetermined mode", and "a synthesis filter section [**560**] which is constituted by spectrum parameters and reproduces a speech signal by filtering the

sound source signal".

The Examiner has stated in the paragraph at the bottom of page 2 of the Office Action mailed January 28, 2004, that "Kleijn et al. do not specifically teach that the discriminating a voice/unvoiced mode is based on a past quantized gain of an adaptive codebook" (emphasis added). First of all, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. The Examiner has further stated in the penultimate paragraph on page 3 of the Office Action that "Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction" (emphasis added). Again, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. In both cases, the Examiner relies on the article by Ozawa et al. to provide a teaching of the missing features and erroneously concludes that the Kleijn et al. coding system could be modified as taught by Ozawa et al. to achieve the claimed invention. The facts are to the contrary, however, since Kleijn et al. and Ozawa et al. are employing different coders based on different algorithms. Modifications of Kleijn et al. as proposed by the Examiner is not fairly taught by the references, and it is not at all clear that such modifications would result in a working coder/decoder. All the Examiner has done is to identify out of context features in Ozawa et al. and say that they could somehow be used to modify Kleijn et al.

As described in the specification, a first characteristic of the present invention is that, after a speech signal is input, a voiced/unvoiced coding mode is discriminated every predetermined frame section by making use of a past quantized gain in the adoptive codebook. By this feature, it is unnecessary to newly calculate a pitch gain, etc., because the past quantized gain is used. Second, it is unnecessary to transmit mode discrimination information to the reception side. Third, therefore, the present

invention can decrease a calculation amount and a transmission amount as compared with the prior art.

A second characteristic feature of the present invention is that, in a search of sound signal source (sound source codebook) consisting of the assemblage of code vectors of non-zero pulse, in a sound source quantization section, the combination of a code vector (pulse) and a shift amount for shifting a position of the code vector (pulse) is searched on the basis of the coding mode. By the second characteristic feature which is not disclosed in either the Ozawa et al. publication or the Kleijn patent, a search coding apparatus of the present invention can increase the degree of freedom of the pulse position and improve remarkably the sound quality in a low bit rate as compared with the prior art.

On the contrary, the Ozawa et al. publication discloses a M-LCELP speech coder in which a coding mode of a predetermined frame is discriminated every predetermined frame section by calculating a pitching gain from an input speech signal. At that time, mode discrimination information must be transmitted to the reception side.

In the speech coding apparatus of the claimed invention, at the time of performing the speech coding, no coding mode information is transmitted. Therefore, in the claimed invention, it becomes possible to code a speech signal efficiently with a relatively small amount of calculation when sound source information is relatively poor and to reduce the transmitting bit rate.

On the contrary, in the prior art, coding mode information is transmitted at the time of performing the speech coding. That is, in the prior art, coding mode information is firstly obtained from a periodicity of inputted speech, then the coding mode information is transmitted. Therefore, in accordance with the coding mode information, such an operation as changing coding mode and/or excitation gain should be done.

For reference, claims 1 and 6 are reproduced below, annotated by bold faced reference numerals taken from drawing Figures 1 and 5:

1. A speech coding apparatus including at least:

a spectrum parameter calculation section **200, 210** for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section **500** for obtaining a delay and a gain from a past quantized sound source signal **366** by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section **370** for discriminating a voiced sound mode and an unvoiced sound mode on a basis of a past quantized gain **366** of an adaptive codebook **380**;

a sound source quantization section **350** which has a codebook **351, 352** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section **370**, and searches combinations of code vectors stored in said codebook **351, 352** and a plurality of shift amounts used to shift positions of the

pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech; and

a multiplexer section **400** for outputting a combination of an output from said spectrum parameter calculation section **210**, an output from said adaptive codebook section **500**, and an output from said sound source quantization section **350**.

6. A speech coding/decoding apparatus comprising:

a speech coding apparatus **(Fig. 1)** including:

a spectrum parameter calculation section **200, 210** for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section **500** for obtaining a delay and a gain from a past quantized sound source signal **366** by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section **350** for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section **370** for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain **366** of a adaptive codebook, and

a codebook **351, 352** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

said sound source quantization section searching combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech, and further including

a multiplexer section **400** for outputting a combination of an output from said spectrum parameter calculation section **210**, an output from said adaptive codebook section **500**, and an output from said sound source quantization section **350**; and

a speech decoding apparatus **(Fig. 5)** including at least:

a demultiplexer section **500** for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section **530** for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section **540** for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information when an output from said discrimination indicates a predetermined mode, and

a synthesis filter section **560** which is constituted by spectrum parameters and reproduces a speech signal by filtering the sound source signal.

The Group 1 claims are separately patentable as directed to the first embodiment shown in Figure 1. In this first embodiment, for voiced sounds, the sound source quantization circuit 350 reads out polarity code vectors from the codebook 351, assigns positions to the respective code vectors, and selects a

combination of code vector and a position which minimizes distortion given by equation (11) on page 24 of the specification. For unvoiced sounds, pulse positions are set at predetermined intervals, and shift amounts for shifting the positions of all pulses are determined in advance. In this case, the sound source quantization circuit 350 receives polarity code vectors from the polarity codebook 352, and searches combinations of all shift amounts and all code vectors to select a combination of a shift amount and a code vector which minimizes distortion given by equation (15) on page 26 of the specification.

The Examiner next grouped claims 2, 5 and 7. Claim 5 has been canceled. Claims 2 and 7, together with claims 9 and 10, comprise the Group 2 claims listed on page 12 of this brief. Claim 2 is an independent claim directed to the second embodiment of the speech encoder shown in Figure 2, and claim 7 is an independent claim directed to the combination of a coding/decoding apparatus employing the coding system of Figure 2 and the decoding system of Figure 5. Claims 9 and 10, which are dependent on claim 8, are directed to the embodiments shown in Figures 2 and 4.

In the middle of page 4 of the Office Action mailed January 28, 2004, the Examiner stated that "Kleijn et al. do not <u>specifically</u> teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive code book" (emphasis added). As already pointed out, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. Again, on page 5 of the Office Action, the Examiner stated that "Kleijn does not <u>specifically</u> teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction" (emphasis added). Once again, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. As before, the Examiner relies on the article by

Ozawa et al. to provide a teaching of the missing features and erroneously concludes that the Kleijn et al. coding system could be modified as taught by Ozawa et al. to achieve the claimed invention. Again, the facts are to the contrary, however, since Kleijn et al. and Ozawa et al. are employing different coders based on different algorithms. Modifications of Kleijn et al. as proposed by the Examiner are not fairly taught by the references, and it is not at all clear that such modifications would result in a working coder/decoder. All the Examiner has done is to identify out of context features in Ozawa et al. and say that they could somehow be used to modify Kleijn et al.

For reference, claims 2 and 7 are reproduced below, annotated by bold faced reference numerals taken from drawing Figures 2 and 5:

2. A speech coding apparatus including at least:

a spectrum parameter calculation section **200, 210** for receiving a speech signal, obtaining a spectrum parameter,

an adaptive codebook section **500** for obtaining a delay and a gain from a past quantized sound source signal **366** by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section **370** for discriminating a voice sound mode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook;

a sound source quantization section **355** which has a

codebook **351, 352** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section **370**, and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses *according to a predetermined rule*; and

a multiplexer section **400** for outputting a combination of an output from said spectrum parameter calculation section **210**, an output from said adaptive codebook section **500**, and an output from said sound source quantization section **355**.

7. A speech coding/decoding apparatus comprising:

a speech coding apparatus **(Fig. 2)** including:

a spectrum parameter calculation section **200, 210** for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section **500** for obtaining a delay and a gain from a past quantized sound source signal **366** by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section **355** for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section **370** for discriminating a voice sound

mode and an unvoiced sound mode on the basis of a past quantized gain **366** of an adaptive codebook, and

a codebook **351, 352** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section,

said sound source quantization section outputting a combination of a code vector and shift amount which minimizes distortion relative to input speech by generating positions of the pulses *according to a predetermined rule*, and further including

a multiplexer section **400** for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus **(Fig. 5)** including at least:

a demultiplexer section **510** for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section **530** for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section **540** for reconstructing a sound source signal by generating positions of pulses *according to a predetermined rule* and generating amplitudes or polarities for the pulses from a code vector when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section **560** which includes spectrum parameters and reproduces a speech signal by filtering the sound

source signal.

The Examiner separately grouped claims 9 and 10, both of which are dependent on claim 8. In the grouping of claims on page 12 of this brief, claims 9 and 10 are included in Groups 2 and 4 claims. Claim 9, which is dependent on claim 8, recites that the sound source quantization means uses a position generated according to a predetermined rule as a pulse position when mode discrimination indicates a predetermined mode. Claim 10, which is dependent on claim 9, recites that, when mode discrimination indicates a predetermined mode, a predetermined number of pulse positions are generated by random number generating means and output to the sound source quantization means. This is shown in the embodiments of Figures 2 and 4. The Examiner cites col. 7, lines 2–26, of Kleijn et al. for a disclosure of the features recited in claims 9 and 10; however, the cited passage does not in fact support the Examiner. What Kleijn et al. describe is calculation of the value of G_{opt} and then ascertaining whether or not the calculated value is greater than a first specified threshold value. This is not what is being claimed.

For reference, claims 9 and 10 are reproduced below, annotated by bold faced reference numerals taken from drawing Figure 2:

> 9. An apparatus according to claim 8, wherein said sound source quantization means uses a position generated *according to a predetermined rule* as a pulse position when mode discrimination indicates a predetermined mode.
>
> 10. An apparatus according to claim 9, wherein when mode discrimination indicates a predetermined mode, a predetermined number of pulse positions are generated by random number generating

means **600** and output to said sound source quantization means.

The Group 2 claims are separately patentable as directed to the embodiment shown in Figure 2. This embodiment differs from the first embodiment, and the claims of Group 1, in the operation of the sound source quantization circuit 355. When voiced/unvoiced discrimination indicates an unvoiced sound, the positions are generated in advance in accordance with a predetermined rule. A random number generating circuit 600 is used to generate a predetermined number of pulse positions which are output to the sound source quantization circuit 355. If the discrimination information from mode discrimination circuit 370 indicates a voiced sound, the sound source quantization circuit 355 operates in the same manner as the sound source quantization circuit 350 in Figure 1. If, on the other hand, the information indicates an unvoiced sound, the amplitudes or polarities of pulses are collectively quantized by using a sound source codebook 352 in correspondence with the positions output from the random number generating circuit 600.

Next, the Examiner grouped claims 3, 8 and 11. Claims 3 and 11 are claims of Group 3 listed on page 12 of this brief. Claim 3 is an independent claim directed to the third embodiment of the speech encoder shown in Figure 3. Claim 8 is also an independent claim directed to a speech coding apparatus, but is listed in Group 5 on page 12 of this brief and is separately treated below. Claim 11 is a dependent claim dependent on claim 8 and recites *inter alia* that, when mode discrimination indicates a predetermined mode, the sound quantization means selects a plurality of combinations from combinations of all code vectors in the codebook and shift amounts for pulse positions in an order in which a predetermined distortion amount is minimized.

In the paragraph bridging pages 5 and 6 of the Office Action, the Examiner stated that "Kleijn et al. do not <u>specifically</u> teach that the discriminating a

voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook" (emphasis added), and at the top of page 7, the Examiner stated that "Kleijn does not specifically teach a multiplexer for the coder or a decoder scheme with demultiplexer and sound source reconstructions" (emphasis added). Once again, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. As before, the Examiner relies on the article by Ozawa et al. to provide a teaching of the missing features and erroneously concludes that the Kleijn et al. coding system could be modified as taught by Ozawa et al. to achieve the claimed invention. Again, the facts are to the contrary, however, since Kleijn et al. and Ozawa et al. are employing different coders based on different algorithms. Modifications of Kleijn et al. as proposed by the Examiner is not fairly taught by the references, and it is not at all clear that such modifications would result in a working coder/decoder. All the Examiner has done is to identify out of context features in Ozawa et al. and say that they could somehow be used to modify Kleijn et al.

For reference, claims 3 and 11 are reproduced below, annotated by bold faced reference numerals taken from drawing Figures 3 and 5:

3. A speech coding apparatus including at least:

a spectrum parameter calculation section **200, 210** for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section **500** for obtaining a delay an a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and

outputting the sound source signal, comprising:

a discrimination section **370** for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain **366** of an adaptive codebook;

a sound source quantization section **356** which has a codebook **351, 352** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based an output from said discrimination section **370**, and a gain codebook **380** for quantizing gains, and searches combinations of code vectors stored in said codebook **351, 352**, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook **380** so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech; and

a multiplexer section **400** for outputting a combination of an output from said spectrum parameter calculation section **210**, an output from said adaptive codebook section **500**, and an output from said sound source quantization section **356**.

11. An apparatus according to claim 8, wherein when mode discrimination indicates a predetermined mode, said sound source

quantization means selects a plurality of combinations from combinations of all code vectors in said codebook **352** and shift amounts for pulse positions in an order in which a predetermined distortion amount is minimized, and outputs the combinations to said gain quantization means, and

said gain quantization means **366** quantized a plurality of sets of outputs from said sound source quantization means by using said gain codebook **380**, and selects a combination of a shift amount, sound source code vector, and gain code vector which minimizes the predetermined distortion amount.

The claims of Group 3 are separately patentable from the claims of Groups 1 and 2 as directed to the embodiment of Figure 3. In this embodiment, when the voice/unvoiced discrimination information from mode discrimination circuit 370 indicates an unvoiced sound, a sound source quantization circuit 356 calculates the distortions given by equations (21) in correspondence with all the combinations of all the code vectors in the sound source codebook 352 and the shift amounts and pulse positions, and outputs them to the gain quantization circuit 366. The gain quantization circuit, in turn, quantizes gains for a plurality of sets of outputs from the sound source quantization circuit 356 by using a codebook 380 and selects a combination of a shift amount, sound source code vector, and gain code vector which minimizes distortions.

The Examiner treated claim 4 as a single group; however, in the grouping of claims on page 12 of this brief, claims 4, 9, 10, and 11 were included in Group 4. Regarding claim 4, this claim recites, "a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal". The sound source quantization section comprises, "a discrimination section [**370**] for discriminating a voice sound mode and

an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook [**500**]", and "a sound source quantization section [**357**] which has a codebook [**351, 352**] for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on output from said discrimination section, and a gain codebook [**380**] for quantizing gains, and outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule".

In the penultimate paragraph on page 7 of the Office Action mailed January 28, 2004, the Examiner stated that "Kleijn et al. do not <u>specifically</u> teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook" (emphasis added), and in the fourth paragraph from the bottom on page 8, the Examiner stated that "Kleijn does not <u>specifically</u> teach a multiplexer for the coder or a decoder scheme with a demultiplexer and sound source reconstruction" (emphasis added). Once again, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. As before, the Examiner relies on the article by Ozawa et al. to provide a teaching of the missing features and erroneously concludes that the Kleijn et al. coding system could be modified as taught by Ozawa et al. to achieve the claimed invention. Again, the facts are to the contrary, however, since Kleijn et al. and Ozawa et al. are employing different coders based on different algorithms. Modifications of Kleijn et al. as proposed by the Examiner is not fairly taught by the references, and it is not at all clear that such modifications would result in a working coder/decoder. All the Examiner has done is to identify out of context features in Ozawa et al. and say that they could somehow be used to modify Kleijn et al.

For reference, claim 4 is reproduced below, annotated by bold faced reference

numerals taken from drawing Figure 4:

4. A speech coding apparatus including at least:

a spectrum parameter calculation section **200, 210** for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section **500** for obtaining a delay an a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section **370** for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain **367** of an adaptive codebook;

a sound source quantization section **357** which has a codebook **351, 352** for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section **370** indicates a predetermined mode, and a gain codebook **380** for quantizing gains, and outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the

pulses according to a predetermined rule; and

a multiplexer section **400** for outputting a combination of an output from said spectrum parameter calculation section **210**, an output from said adaptive codebook section **500**, and an output from said sound source quantization section **357**.

Claims 9 and 10 were reproduced above in connection with the second embodiment of and Group 2 claims, but are included in Group 4 claims as being generic to the embodiments of Figures 2 and 4. Claim 11 was reproduced above in connection with the third embodiment and Group 3 claims, but is included in Group 4 claims as being generic to the embodiments of Figures 3 and 4. The claims of Group 4 are separate patentable over the claims of Groups 1, 2 and 3 as directed to the embodiment of Figure 4. In this embodiment, when the voiced/unvoiced discrimination information from the mode discrimination circuit 370 indicates an unvoiced sound, the sound source quantization circuit 357 collectively quantizes the amplitudes or polarities of pulses for the pulse positions generated by a random number generating circuit 600 by using a sound source codebook 352 and outputs all the code vectors or a plurality of code vector candidates to the gain quantization circuit 367. The gain quantization circuit 367 quantizes gains for the respective candidates output from the sound source quantization circuit 357 by using a gain codebook 380 and outputs a combination of a code vector and gain code vector which minimizes distortion.

Claim 8 is separately listed as Group 5 on page 12 of this brief. Claim 8 is generic to the four embodiments of Figures 1, 2, 3, and 4 and is directed to a speech coding apparatus which comprises, *inter alia*, "mode discrimination means [**370**] for receiving a past quantized adaptive codebook gain and performing mode discrimination associated with a voiced/unvoiced mode by comparing the gain with a

predetermined threshold" and "sound source quantization means [**350, 355, 356, 357**] for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech".

In his discussion of claim 8, at the paragraph bridging pages 5 and 6 of the Office Action, the Examiner stated that "Kleijn et al. do not <u>specifically</u> teach that the discriminating a voiced/unvoiced mode is based on a past quantized gain of an adaptive codebook" (emphasis added), and at the top of page 7, the Examiner stated that "Kleijn does not <u>specifically</u> teach a multiplexer for the coder or a decoder scheme with demultiplexer and sound source reconstructions" (emphasis added). Once again, the use of the adverb "specifically" is misleading, suggesting that Kleijn et al. might *implicitly* teach this feature, when in fact there is no suggestion whatsoever of this feature. As before, the Examiner relies on the article by Ozawa et al. to provide a teaching of the missing features and erroneously concludes that the Kleijn et al. coding system could be modified as taught by Ozawa et al. to achieve the claimed invention. Again, the facts are to the contrary, however, since Kleijn et al. and Ozawa et al. are employing different coders based on different algorithms. Modifications of Kleijn et al. as proposed by the Examiner is not fairly taught by the references, and it is not at all clear that such modifications would result in a working coder/decoder. All the Examiner has done is to identify out of context features in Ozawa et al. and say that they could somehow be used to modify Kleijn et al.

For reference, claim 8 is reproduced below, annotated in bold face type with the reference numerals from Figures 1, 2, 3, and 4:

8. A speech coding apparatus comprising:

a spectrum parameter calculation section **200, 210** for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter;

means for obtaining a delay and a gain from a past quantized sound source signal **366, 360** by using an adaptive codebook, and obtaining a residue **240, 235** by predicting a speech signal; and

mode discrimination means **370** for receiving a past quantized adaptive codebook gain **366** and performing mode discrimination associated with a voiced/unvoiced mode by comparing the gain with a predetermined threshold, and

further comprising:

sound source quantization means **350 [355, 356, 357]** for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook **351, 352** for collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech;

gain quantization means **366** for quantizing a gain by using a gain codebook **380**; and

multiplex means **400** for outputting a combination of outputs from said spectrum parameter calculation means **210**, said adaptive codebook means **500**, said sound source quantization means **350**, and said gain quantization means **366**.

Claim 8, the claim of Group 5, is separately patentable over the claims of Groups 1, 2, 3, and 4 as being generic to the four embodiments shown in Figures 1, 2, 3, and 4.

In his Advisory Action mailed August 9, 2004, the Examiner stated that he "recognizes that obviousness can *only* be established by combining or modifying the teachings of the prior art to produce the claimed invention where there is some *teaching, suggestion*, or *motivation to do so found in the references themselves* or *in the knowledge generally available to one of ordinary skill in the art*" (emphasis added, citations omitted). Having said that, the Examiner abandoned all effort to demonstrate that there is "some *teaching, suggestion*, or *motivation to do so found in the references themselves*", since there is in fact none, but relied instead on some notion of "*knowledge generally available to one of ordinary skill in the art*", without demonstrable evidence supporting his position. In the Request for Reconsideration Under 37 C.F.R. §1.116 filed on July 28, 2004, the Examiner was asked whether he knew of such evidence in the form of a publication or issued patent, and if he did, he was requested to cite it. The Examiner was asked if he was relying on his own knowledge, and if so, he was requested to provide an affidavit stating the nature of his knowledge, when he acquired it, and under what circumstances. In his Advisory Action mailed on August 8, 2004, the Examiner failed to respond by providing either the citation of such a publication or issued patent or the submission of a probative affidavit by the Examiner. The rejection of the claims can only be viewed as a subjective application of the standards under Section 103 based on hindsight, whereas

the appropriate standard is an objective standard. The rejection should therefore be reversed.

ARGUMENT VIIIE. REJECTION OTHER THAN 35 U.S.C. §§102, 103 AND 112

There are no other rejections other than the rejection under 35 U.S.C. §103.

IX. APPENDIX OF CLAIMS INVOLVED IN THE APPEAL (37 C.F.R. §1.192(C)(9))

The text of the claims involved in the appeal are:

1. A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voiced sound mode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

2. A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on a basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section, and outputs a code vector that minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

3. A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay an a gain from a past quantized sound source signal by using an adaptive codebook, and

obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based an output from said discrimination section, and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

4. A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay an a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source

signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section indicates a predetermined mode, and a gain codebook for quantizing gains, and outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

6. A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

said sound source quantization section searching combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information when an output from said discrimination indicates a predetermined mode, and

a synthesis filter section which is constituted by spectrum parameters

and reproduces a speech signal by filtering the sound source signal.

7. A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of an adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on an output from said discrimination section,

said sound source quantization section outputting a combination of a code vector and shift amount which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating positions of pulses according to a predetermined rule and generating amplitudes or polarities for the pulses from a code vector when an output from said discrimination section indicates a predetermined mode, and

a synthesis filter section which includes spectrum parameters and reproduces a speech signal by filtering the sound source signal.

8. A speech coding apparatus comprising:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter;

means for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal; and

mode discrimination means for receiving a past quantized adaptive codebook gain and performing mode discrimination associated with a voiced/unvoiced mode by comparing the gain with a predetermined threshold, and

further comprising:

sound source quantization means for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the signal, and searching combinations of code vectors stored in a codebook for

collectively quantizing amplitudes or polarities of a plurality of pulses in a predetermined mode and a plurality of shift amounts used to temporally shift a predetermined pulse position so as to select a combination of an index of a code vector and a shift amount which minimizes distortion relative to input speech;

gain quantization means for quantizing a gain by using a gain codebook; and

multiplex means for outputting a combination of outputs from said spectrum parameter calculation means, said adaptive codebook means, said sound source quantization means, and said gain quantization means.

9. An apparatus according to claim 8, wherein said sound source quantization means uses a position generated according to a predetermined rule as a pulse position when mode discrimination indicates a predetermined mode.

10. An apparatus according to claim 9, wherein when mode discrimination indicates a predetermined mode, a predetermined number of pulse positions are generated by random number generating means and output to said sound source quantization means.

11. An apparatus according to claim 8, wherein when mode discrimination indicates a predetermined mode, said sound source quantization means selects a plurality of combinations from combinations of all code vectors in said codebook and shift amounts for pulse positions in an order in which a predetermined distortion amount is minimized, and outputs the combinations to said gain quantization means, and

said gain quantization means quantized a plurality of sets of outputs

from said sound source quantization means by using said gain codebook, and selects a combination of a shift amount, sound source code vector, and gain code vector which minimizes the predetermined distortion amount.

X. OTHER MATERIALS THAT APPELLANT CONSIDERS NECESSARY OR DESIRABLE

There are no other materials that appellant considers necessary to desirable to decide this appeal.

Respectfully submitted,

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